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A RATIONALE FOR CODING
ON A SATELLITE PACKET BROADCASTING CHANNEL

1. Purpose

The purpose of this study is to provide a systematic framework for the investigation and synthesis of coding techniques which will allow efficient use of both small and large earth stations operating in a broadcast packet satellite network. This study is related particularly to the channel error characteristics of the ATS-1 VHF channel, which is quite noisy in comparison to typical SHF satellite communication channels, and therefore represents a worse-case condition.

2. Objective

The objective of this study is to produce efficient coding designs providing error control, packet synchronization, and header identification in the presence of noise for a mix of small and large earth stations operating in a broadcast packet satellite network. The designs will be applicable to the measured error characteristics of the ATS-1 VHF channel, for use in future experiments needed to verify the designs.

3. Channel Characterization

Several basic assumptions may be used to characterize the ATS-1 VHF satellite channel when used for packet broadcasting in the pure-ALOHA random-

access mode. These are:

(1) It is a multiaccess channel.

(2) Packet conflicts will occur at the transponder according to statistics described by Abramson [1].

(3) Packet lengths are either fixed or variable.

(4) Noise and interference exist in the channel.

(5) A feedback channel with delay exists, subjected to noise and interference similar to that in the forward channel.

(6) The channel may be described in terms of bit error patterns, assuming that bit synchronization is performed in a conventional manner.

The first assumption results from the very nature of packet broadcasting and resource sharing. That is, a common satellite transponder can be accessed by a multiplicity of earth stations, each time-sharing the common channel with all the other stations.

The second assumption results from operation of the multiaccess channel in an ALOHA random-access mode, wherein each transmitter attempts to access the common channel at uncontrolled times. This mode of operation results in finite probabilities of packet conflicts at the common transponder.

The third assumption simply states that both fixed- and variable-length packet operational modes should be considered in the study. Variable length adds another dimension to the packet detection process.

The fourth assumption is a general one which can be made for many communication channels, and the ATS-1 VHF channel is no exception. The transfer of digital data via satellite communication channels encounters the universal problem of errors introduced at the receiver by various forms of channel noise. The remarkable characteristic of the SHF space channel is that it

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appears to be accurately modeled as a white-Gaussian noise channel, and therefore lends itself to treatment by the most sophisticated and powerful coding procedures. Unfortunately, the VHF space channel is not quite so tidy, particularly in an urban environment, due to the difficulty of attaining sufficient antenna directivity to screen out man-made noise sources, coupled with the higher intensity levels of such noise at the lower frequencies. In addition, co-channel interference is much more likely, due to the large number of emitters operating at VHF. The result of these conditions is to produce a bit-error characteristic in the channel which is a mix of both random and burst errors, which is also time-varying in nature. Such is the characteristic of the ATS-1 VHF satellite communication channel which has been dealt with experimentally by THE ALOHA SYSTEM. (Examples of error rate magnitude and variability for this channel are shown in Tables 1 through 3 of reference [2].)

Assumption five stems from the condition that any receiving earth station can communicate back to the transmitting earth station via the broadcast channel. A round-trip delay is incurred in using this feedback channel, which can have serious consequences on throughput, depending upon the type of error control method used. The noise and interference characteristics for the forward and feedback channels are not necessarily the same, particularly in the VHF channel where the local noise and interference at each earth station may be different.

The final assumption is based on the use of conventional narrow-band bit-synchronous data transmission. For the experimental system considered, carrier-synchronization is not a factor since frequency modulation of the RF

carrier by the data signal is used.

4. Signal Detection

The general problem addressed by possible coding strategies for the channel described in the previous section is the detection of a signal of unknown arrival time, length, and perhaps shape, in the presence of noise and interference. For optimal detection, a maximum likelihood detector should be used. However, because of the almost infinite number of possible signal combinations, it is impossible to directly use the maximum likelihood approach. The use of a sequential detection procedure reduces the number of possibilities exponentially. For broadcast packet detection in the multiaccess channel, the sequence followed is:

(1) Carrier Sync

(2) Bit Sync

(3) Frame Sync

(4) Packet Identification

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(5) Error Detection/Correction

(6) Signal Termination

Processes (1) through (4) taken together constitute what is known as Packet Acquisition, i.e., the detection of the existence of a signal. Process (5) is the function provided by error control schemes. Process (6) is necessary to reset the receiver for processing of the next-arriving packet and terminate the data stream into the data processing equipment. Since carrier sync is not necessary in the ATS-1 experimental system and bit sync is performed in a conventional manner, we need only consider the problems of frame sync,

identification, error control, and signal termination. The last item, signal termination, may be incorporated as a subset of the frame sync problem, since the packet length is known at the receiver a priori for fixed-length packets, and may be encoded into the packet header information for variable-length packets. Therefore, our basic consideration of the packet detection problem involves the interaction of the processes of frame sync, identification, and error control.

Frame sync and packet identification are related since the ID bit pattern for any given earth station is fixed for all packets addressed to that station, and this signal redundancy may be used to advantage in the frame synchronizing process. Therefore, frame sync and identification will be considered together as a combined process, called acquisition, and error control will be examined as the other primary detection process.

5. The Acquisition Problem

The acquisition system is the front end of a user terminal control unit (Fig. 1), and serves the purpose of synchronizing received packets and determining which packets belong to the local terminal.

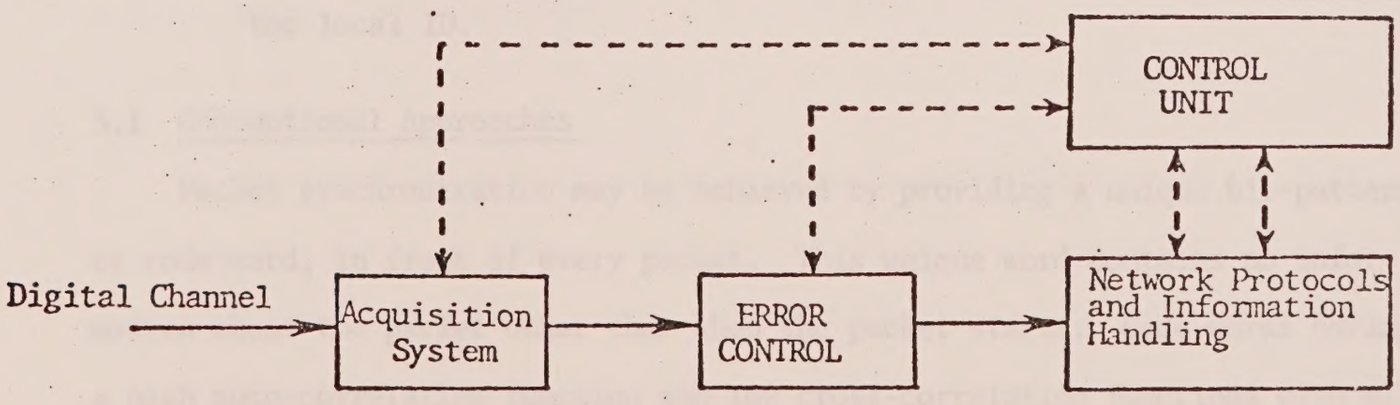

FIG. 1 TERMINAL CONTROL UNIT

A multi-terminal environment is assumed in which packets belonging to all terminals are transmitted on one frequency. Each packet carries a destination ID which allows a receiving terminal to filter out packets meant for other destinations.

The input to the acquisition system consists of an infinite train of bits with packets from various sources distributed at random. This distribution of packets in time complicates the acquisition process--arrival times are not periodic and frame sync must be established each time a packet is received. (We assume bit sync is attained, using an appropriate bit-sync preamble.) In addition to channel noise, packet collisions may occur where two or more packets overlap in time, destroying all or part of the information in the involved packets.

In this environment, packet acquisition involves the following steps:

(i) Packet synchronization. This is essentially the determination of where a packet starts in time.

(ii) Determination of packet length. This step would be necessary only in a variable packet length environment. (Packet length along with (i) completes the information needed to fix packet boundaries.)

(iii) Determination of packet destination and selection of packets bearing the local ID.

5.1 Conventional Approaches

Packet synchronization may be achieved by providing a unique bit-pattern, or code-word, in front of every packet. This unique word contains no information about the packet other than when the packet starts. Code-words having a high auto-correlation function and low cross-correlation functions with all other equal-length code-words are usually selected for the unique word. This

is the most reliable synchronization approach since all receivers need only look for a single unique code-word in order to frame the packet. However this does little to help identify a packet intended for a particular receiver. Length and ID information have to be provided in a separate word in a coded form to protect against errors, and consequently an additional decoding step is required once synchronization is attained.

One possible approach is to find a set of code-words each having a maximum ratio of auto-correlation-to-cross-correlation among themselves, and use these code-words for both synchronization and identification purposes. No general theory seems to exist that directly determines the structure of such code-words, so that a computer search becomes necessary to find code-words which have this property. Code-words satisfying this requirement tend to increase in length much more rapidly than the rate of increase in the number of addresses, so that for any system employing more than just a few addresses these code-words become very lengthy [3]. A trade-off word-length is soon reached where this code-word approach becomes less efficient than the use of a combination of separate sync-word and address-word. This is due largely to the advantage of employing sequential decisions, where decision error probabilities are considered independent and thus multiply together to give a lower final probability for a given total number of bits.

5.2 Observations

In the acquisition process, two distinct types of errors are possible:

A *miss error* occurs when a packet with the correct identification is in sync, but acquisition fails because of severe errors in the sync-word.

A *false-alarm error* occurs when a packet is not in sync, but an arbitrary bit sequence causes acquisition to be indicated.

The basic issues involved in the packet acquisition process are the minimization of miss probability and false-alarm probability for a given level of encoding redundancy, or conversely, the minimum amount of encoding redundancy to obtain given maximum miss and false-alarm probabilities. The latter requirement is directed toward maximizing channel throughput and consists of a trade-off of the miss and false alarm probabilities against the amount of added redundancy to provide a maximum overall system throughput. This trade-off will be strongly influenced by the type of communication channel used. If the channel is not orthogonal, as exemplified by the ALOHA channel, then minimization of packet length becomes a dominant factor, and the ALOHA packet throughput equations must be considered. In this case maximum miss and false-alarm rates are selected which will cause the packet acquisition statistics to have a secondary effect on the random-access channel statistics. With this approach, the random-access channel statistics remain dominant.

The conventional approaches discussed previously provide sufficient encoding redundancy at the beginning of a packet to enable reliable frame-sync. They do not provide information on packet length (for variable-length packets) or any additional information used for control purposes. This information is contained in the packet header following the sync word in a coded form to protect against errors as discussed previously. Therefore, the conventional approach appears somewhat inefficient in the use of redundancy for frame-sync and error-control of control information.

As an alternative approach, packet destination, length and possibly protocol information can be encoded using a linear block code and the code-word, put in front of every packet, used for frame-sync as well as transmitting information.

This method makes more efficient use of the redundant bits added in the encoding process and should tend to minimize total packet length. However, because of the dual functions of frame-sync and error-control served, this approach suffers some loss of redundancy compared to the conventional approaches mentioned and therefore will exhibit higher miss and false-alarm rates. If these miss and false-alarm probabilities are acceptable for system operation, this approach can be very attractive, for reasons to be discussed in a later section.

6. The Error Control Problem

As discussed previously, the ATS-1 VHF channel exhibits a bit-error characteristic which is a mix of both random and burst errors, varying in average intensity with time. Therefore the channel tends to enter into "noisy" modes and "quiet" modes in a random manner. The bit-error rate averages about 1×10^{-3}, but can deviate considerably from this average as shown in the tables from reference [2]. Measurements of packet throughput versus bit errors per packet have been made by transmitting packets over the channel. Fig. 2 is the result of tests made with the channel in a "quiet" mode, for three different data rates. The number of errors per packet tend to be exponentially distributed, with the largest number of erroneous packets containing two error bits. (Even numbers of errors are predominant in the statistics because of the differential encoding used in the ALOHA modem.) The shape of the distribution is similar when the channel is in its "noisy" mode, with the primary difference being fewer error-free packets and a longer "tail" to the distribution. That is, the distribution retains its exponential characteristic, but exhibits a higher average bit-error rate.

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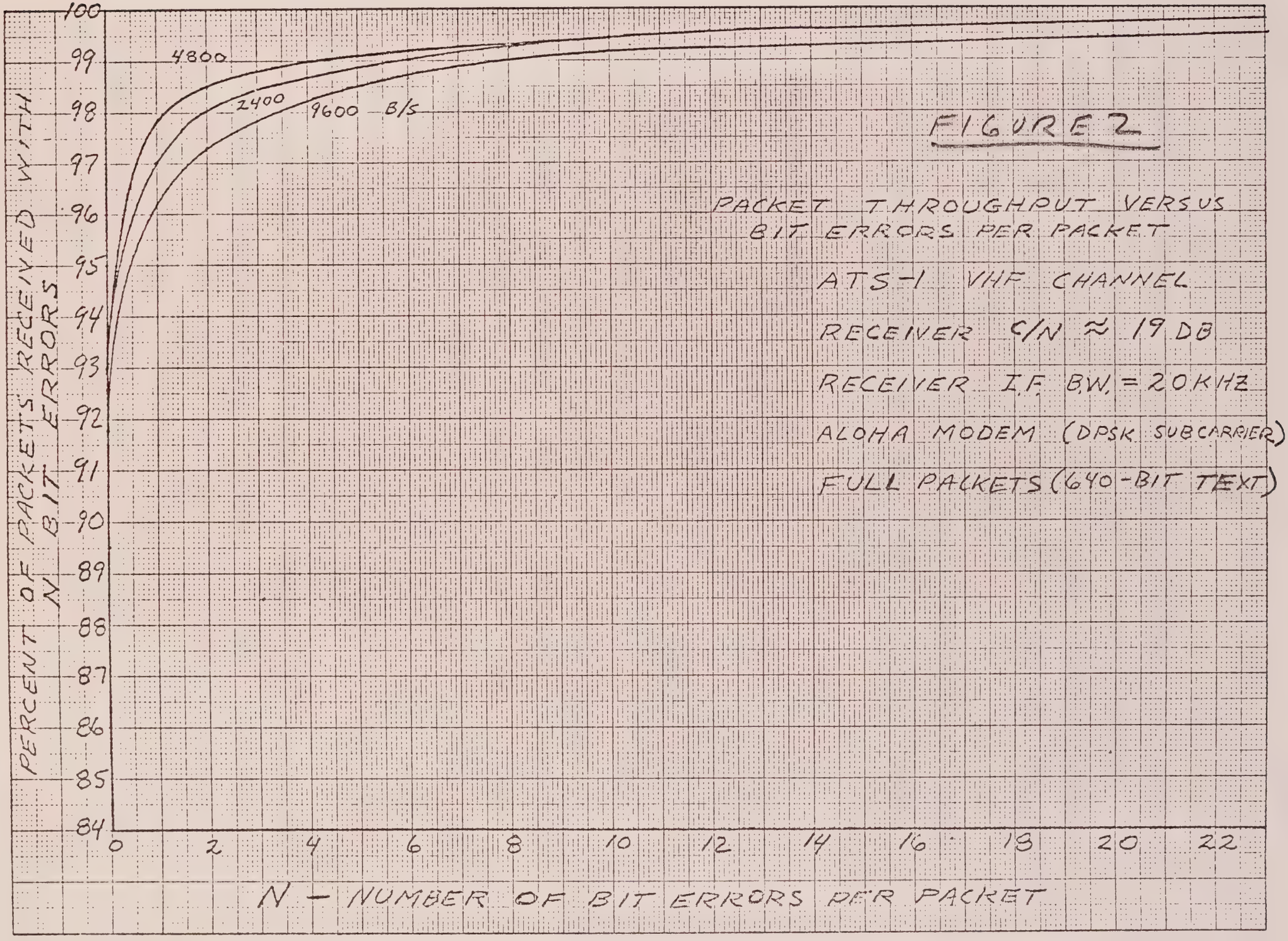

FIGURE 2

PACKET THROUGHPUT VERSUS BIT ERRORS PER PACKET

ATS-1 VHF CHANNEL

RECEIVER C/N ≈ 19 DB

RECEIVER I.F. B.W. = 20 KHZ

ALOHA MODEM (DPSK SUBCARRIER)

FULL PACKETS (640-BIT TEXT)

In addition to the bit errors caused by noise, errors are also caused by packet conflicts due to the random accessing of the satellite transponder by earth stations. This ALOHA packet broadcasting mode of operation results in packets which can have long error bursts associated with them. Some long-duration noise bursts will also produce this same effect.

The error-control problem is therefore how to maximize system throughput (at the cost of some given delay) under the above given conditions of channel errors. First we will discuss conventional approaches.

6.1 Conventional Approaches

When digital data is to be transmitted from one point to another in a noisy environment, some sort of error control mechanism is usually required to keep error rates below a tolerable level. There are two basically different classes of techniques which are used for this purpose. The first one, called forward error correction (FEC), involves encoding messages into longer codewords at the transmitter such that the redundant part can be used at the receiver to determine the pattern of errors that occurred. The decoder (Fig. 3) computes the redundant bits from the received message and compares them to the received redundant bits. The difference contains information about the error pattern that permits the receiver to correct the errors. The reverse channel shown in Fig. 3 is not used in FEC.

The other class of techniques is known as Automatic-Repeat-Request (ARQ). Here, redundancy is added to messages as in FEC, but the decoder only tries to determine if errors have occurred, making no attempt to identify the pattern. If errors have occurred, the receiver uses a reverse channel (Fig. 3) to signal a negative acknowledgement (NACK) to the transmitter, causing it to retransmit the message. If the receiver determines that no errors have

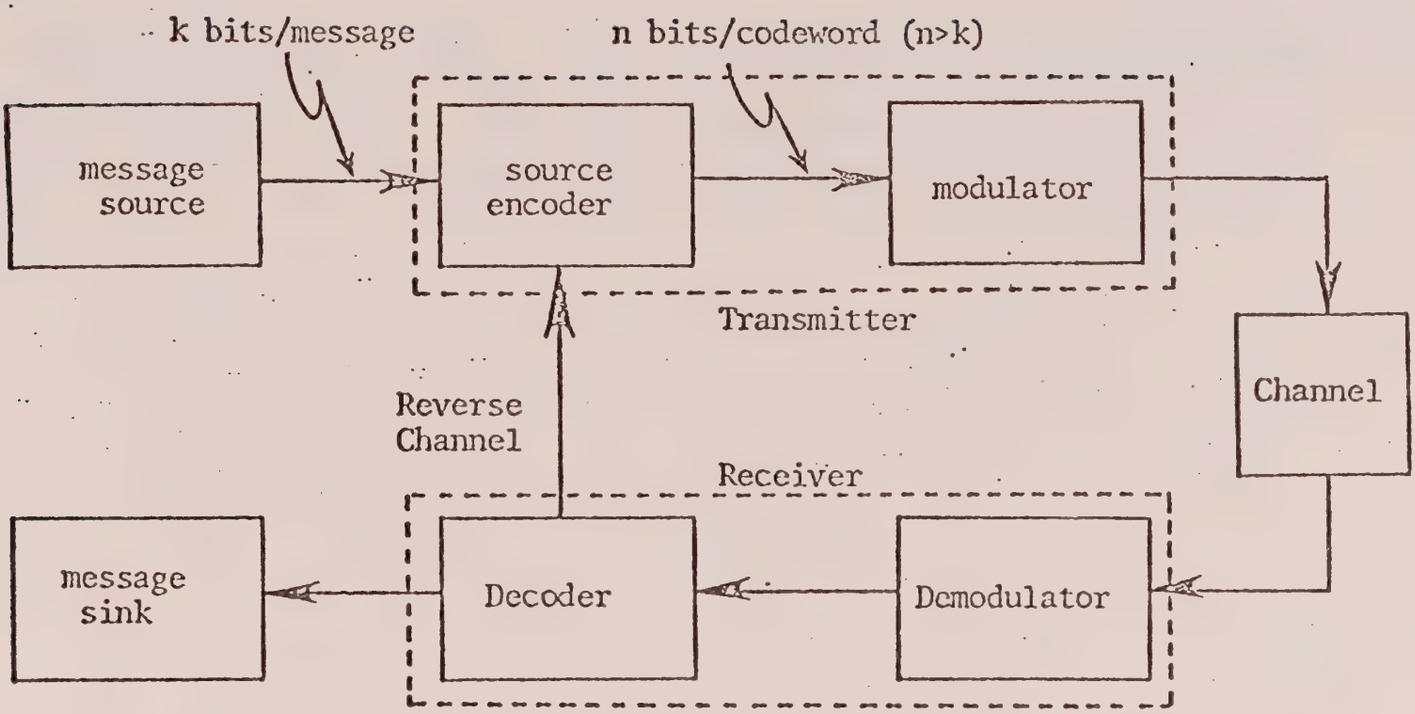

Figure 3. A General Error Control System. (The reverse channel is not used when the system is in FEC mode.)

occurred, it sends a POSACK which inhibits the retransmission. In one variation of ARQ, used on THE ALOHA SYSTEM, only POSACKS are sent, the message being retransmitted automatically after a time if no POSACK has been received. All ARQ systems require the message to be buffered at the transmitter till a POSACK is received.

It has been found that for many real digital data communication channels, high rate FEC schemes that can be simply implemented do not result in significant improvement in error rates. The reason for this is that FEC works well when almost all the errors that occur on the channel form a small fraction of all possible error patterns. This condition is satisfied, for example, in a channel where errors are independent events occurring with a probability $p \ll 1$. Here, random error correcting codes work very well even when the code used has

a small correcting capability. If, on the other hand, errors have a well-defined bursty nature so that almost all patterns that occur are confined to a span of B bits, where B is reasonable, a burst error correcting code can be used very effectively.

Unfortunately, available channels (the ATS-1 channel being no exception) do not satisfy these conditions to any appreciable extent, so that FEC schemes that implement easily do not work well. Coding schemes have been developed for channels with composite burst and random errors, but here either the decoding algorithm is complicated or the rate of the code is low.

ARQ schemes on the other hand are much simpler to implement since only error detection needs to be done. The probability of error in ARQ is determined by the probability of an error going undetected, and this number is of the order of 2^{-p}, where p is the number of parity bits, independent of the nature of errors on the channel. This insensitivity of ARQ makes it very attractive for use on channels with time dependent error characteristics.

6.2 Observations

Previous studies on this channel using a rate 1/2 convolutional code for error correction showed that significant improvement can be made in error rates [4]. However, after examining the time varying nature of errors and the fact that the convolutional code worked even under worst case conditions, it was felt that a comparable improvement in error rates could be obtained using higher-rate block codes,and thus improve overall channel throughput. Block codes lend themselves more naturally to the burst nature of packet communications, and, more importantly, to adaptive schemes which can adjust to the time-varying error rate of the channel. The basic thinking is that overhead would be determined by channel conditions, so that average throughput would be greater than

that provided by any fixed scheme, for the same improvement in error probability. Ideally, such schemes would lend themselves to channels having very different error characteristics, so that with slight modifications they could be universally used for both large and small ground stations.

7. Proposed Design Approach

7.1 Packet Acquisition

As discussed previously, packet destination, length and possibly protocol information can be encoded using a linear block code and the code-word, put in front of every packet, used for synchronization as well as transmitting information. This method is attractive for a number of reasons:

(i) One intuitively feels that the overhead is smaller since the same word is used for two purposes.

(ii) The number of distinct decoding steps in the acquisition process is reduced.

(iii) Decoder complexity compares well with decoders using correlation detection.

(iv) The current ALOHA packet format can be retained.

Since the channel's error-rate is time-varying and can get very noisy at times, the miss probability can get high enough at these times to significantly effect the channel throughput statistics. With correction of just short single bursts in the sync-word, the miss probability can be kept low enough so that the random-access conflict statistics remain dominant. However, the penalty for incorporating this error-correction capability in the sync-word is a higher false-alarm probability.

When operating in a noisy environment, any good acquisition scheme would have to trade-off the probabilities of miss error and false-alarm error. False alarm rates can be kept low by not doing any correction on the sync-word. However, this would--in the worst case-- limit packet throughput at the acquisition stage to about 75%. With correction of just short single bursts in the sync-word, throughput rises significantly when the channel is in its noisy mode and marginally when in its quiet mode. Burst error correction can be mechanized with little added complexity, and if the system can tolerate the increased false alarm rate, error correction on the sync-word should be incorporated. For a synchronization scheme based on a linear block code to be practical the decoder must be simple and *must be able to decode words (including error correction) at the bit rate of the channel.* The first constraint almost immediately narrows down the search for a code to cyclic codes, which is among the best studied and most easily implemented class of linear block codes. The second constraint presents considerable problems even at moderately high data rates unless the conventional decoding scheme is modified. A recursive decoding algorithm can be used which makes use of information about the previously decoded word to decode the current word.

The major blocks of the acquisition system are shown in Fig. 4. The received bit sequence is continuously shifted into an ℓ-bit buffer register. After every bit shift, the sync detection circuit examines the contents of this register to see if it can be corrected into a codeword. When a correctable pattern is found, the circuit indicates sync and simultaneously puts out the error pattern. An ID match circuit compares the corrected upper q bits of the buffer register with the local ID. When a sync and a match occur together, acquisition is indicated.

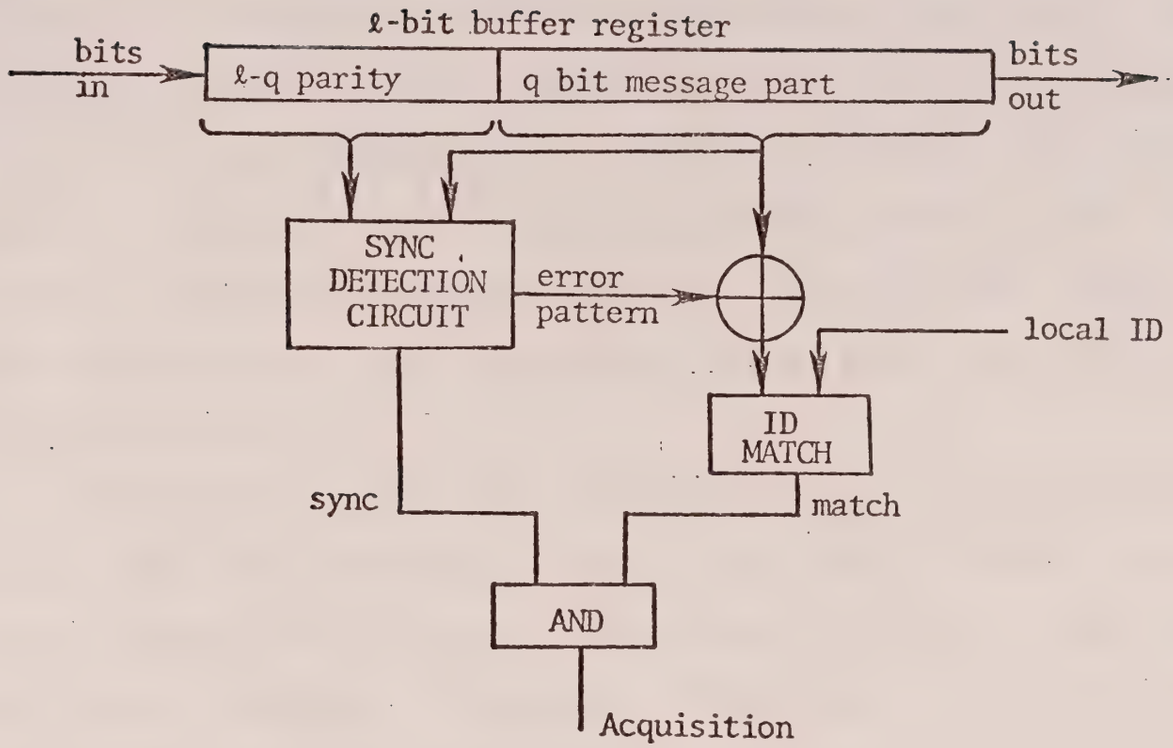

FIG. 4 MAJOR BLOCKS OF THE ACQUISITION SYSTEM

7.2 Error Control

In spite of their simplicity and inherent insensitivity, pure ARQ schemes suffer from inefficient use of the channel especially when multiple retransmits are required for a particular message. Hybrid error control schemes which have an FEC capability embedded inside ARQ get around this problem to some extent. The question that naturally arises is whether it is possible to improve on hybrid ARQ-FEC by cutting down the average number of retransmits even further. This can indeed be done. The basic idea behind the improvement is that discarding a packet which is found to have errors, is throwing away valuable information -- information that could well be used in conjunction with a retransmit to correct many more errors than would otherwise be possible. If decoding based on two copies fails, another retransmit is called for and

correction is attempted using the cumulative store of three copies. This process continues till either a correct packet is received or it is possible to decode the message from the copies. It can be seen heuristically, that the number of retransmits required would be reduced considerably if the redundancy present in the form of message multiplicity is carefully used. Such a system will work much better with a time varying channel than either FEC, pure ARQ or hybrid ARQ-FEC.

Consider, for example, what happens when each of these schemes is used on a channel which periodically enters a 'noisy mode'. For such a channel, errors are very likely to occur in the retransmits during the noisy mode. FEC will fail completely if the errors are bad enough. ARQ will keep requesting retransmits, finally giving up when too many retransmits have occurred. Hybrid ARQ-FEC will not work any better unless the inbuilt FEC is able to correct at least one of the copies. The extended ARQ scheme described above has a better chance of being able to correct such errors because of the large redundancy present.

When the distribution of errors per packet is examined (Fig. 5), it is noticed that double adjacent errors account for around 40% of all packets in error. The long tail of this curve is the single factor which prevents forward error correction from working well on this channel. Consider the improvement that results when a random error correcting code with correcting ability t is used. From the cumulative figures in Fig. 5, 40% of the errors remain uncorrected when t=8, 33% remains uncorrected with t=12, and 29% with t=16. It is clear that the largest gains are to be obtained by correcting just two or four errors. Any further improvement using FEC is marginal as long as we

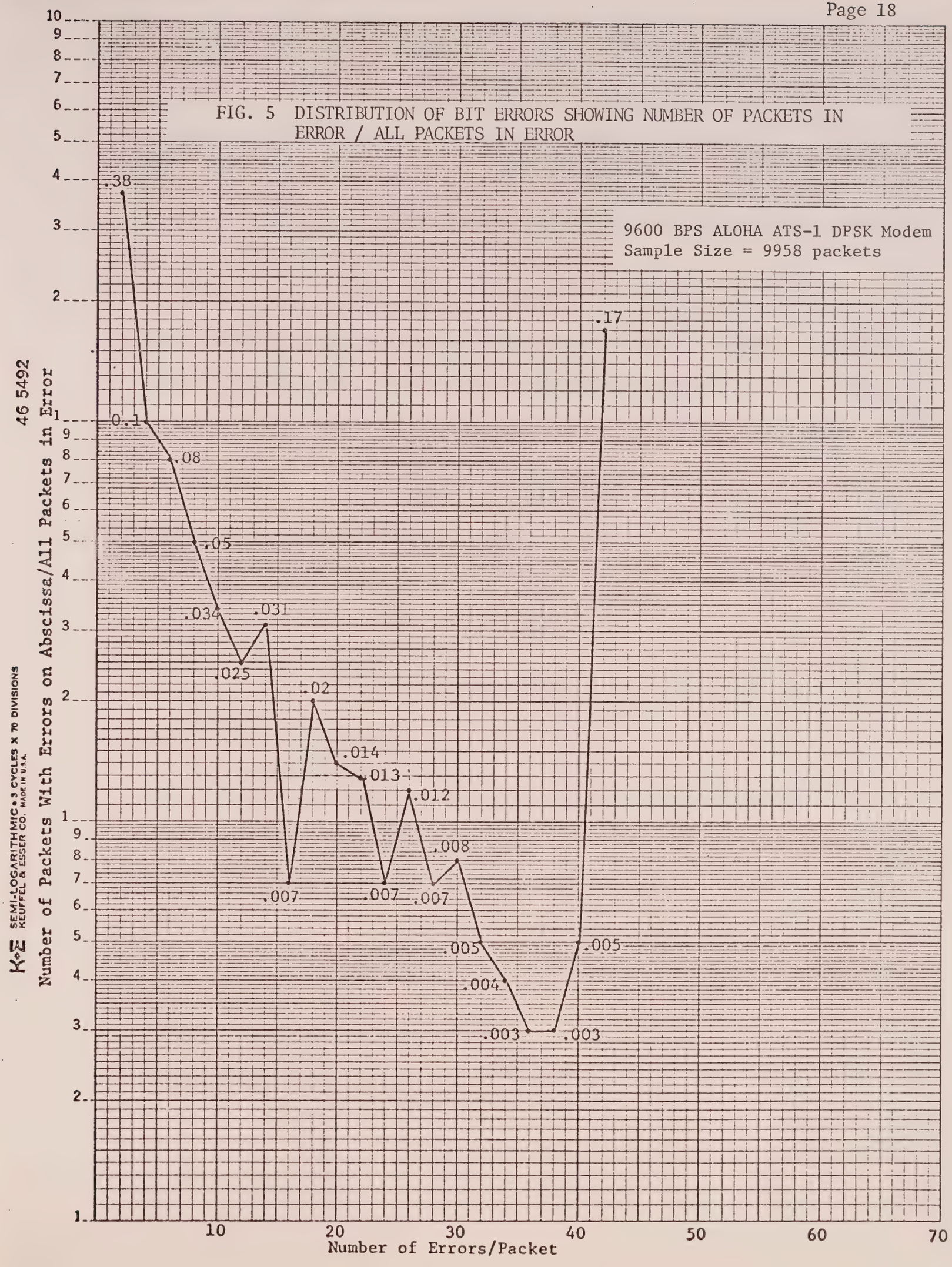
FIG. 5 DISTRIBUTION OF BIT ERRORS SHOWING NUMBER OF PACKETS IN ERROR / ALL PACKETS IN ERROR
9600 BPS ALOHA ATS-1 DPSK Modem
Sample Size = 9958 packets
Number of Packets With Errors on Abscissa/All Packets in Error
Number of Errors/Packet
.38
0.1
.08
.05
.034
.031
.025
.02
.014
.013
.012
.008
.007
.007
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restrict ourselves to high rate codes and simple implementation. If correction based on repetition is used, it is estimated that with a single repeat around 40 errors occurring in one of the packets can be corrected, accounting for all but 17% of the packets in error.

Since significant improvement results from correcting double adjacent errors using FEC, a hybrid system which uses repetition correction with an embedded FEC will work better than a pure repetition system. We call this scheme extended hybrid ARQ-FEC. The throughput T defined as

$$T = \frac{\text{Number of Message Bits}}{\text{Number of Channel Bits}}$$

will now be calculated for this scheme.

Assume a message length of ∿700 bits and a parity of 24 bits* for the embedded FEC. The throughput will be given by

$$T = \frac{n_m}{(n_m+n_p)+f(n_m+n_p)}$$

where n_m = number of message bits
n_p = number of parity bits
f = fraction of all packets uncorrected by the embedded FEC

This formula assumes that all errors can be corrected with a single repetition, an assumption that introduces only second order effects. From Fig. 6 we have $f \approx 0.1$

$$\therefore T = \frac{700}{724+72\cdot4} \simeq \frac{7}{8} \simeq 0\cdot88.$$

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*Correction of single paired errors while maintaining a good detection capability, for ARQ to work well, will require around 24 bits.

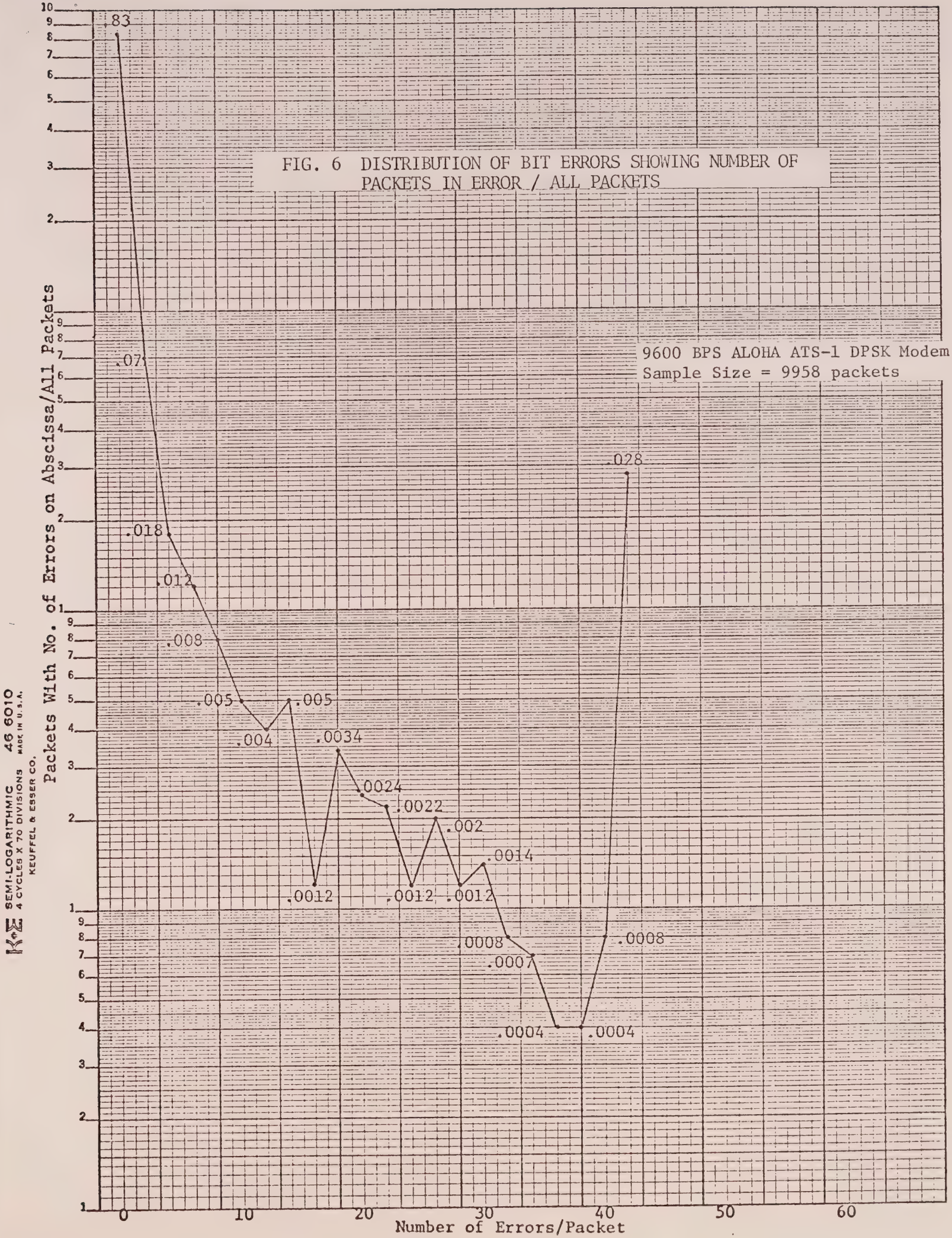

FIG. 6 DISTRIBUTION OF BIT ERRORS SHOWING NUMBER OF PACKETS IN ERROR / ALL PACKETS

For comparison, consider the correcting capability of a pure FEC system that gives the same throughput. The best known constructive codes, the BCH codes, give an error correction rate of 0.015 [5], which is around 11 bits out of 700. For t=12, 33% of the errors remain uncorrected (previous discussion).

The arguments presented above give the rationale for choosing the extended hybrid ARQ-FEC scheme. No claim is made here about the optimality of correction by repeating the message verbatim. It may be argued, for example, that this is an inefficient way of using the repeat packet and that more powerful methods might exist where the repeated packet would be some encoded version of the first packet. Such schemes would probably perform better, but whether the added complexity both in decoding and at the transmitter is worth the gains is an open question. Note that if one insists that the message be recoverable from a single error-free packet (a quite reasonable requirement), that this fixes a minimum rate for the code and will make repetition probably close to optimal.

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[5]